International Civil Aviation Organization

AERONAUTICAL TELECOMMUNICATION NETWORK IMPLEMENTATION COORDINATION GROUP – EIGHTH WORKING GROUP MEETING (ATNICG WG/8)

Christchurch New Zealand
28 September – 1 October 2010

Agenda Item 10: VoIP Strategy for the Region - Guidance

VOICE OVER INTERNET PROTOCOL (VOIP) INTEGRATION TO AIR TRAFFIC SERVICE (ATS) VOICE COMMUNICATION

(Presented by United States of America)

SUMMARY

The existing Air Traffic Service (ATS) voice communication has been used to coordinate air traffic transfer between adjacent FIRs. Even though, Air Traffic Service Inter-facility Data Communication (AIDC) has been in used for more efficient coordination, the ATS voice remains a critical part in day to day air traffic coordination between States' adjacent FIRs. The current voice communication is based largely on analog signaling using a voice grade circuit or multiplexing technique to combine more than one voice channels into a voice grade circuit. The International Civil Aviation Organization (ICAO) has been considering the use of Voice over Internet Protocol (VoIP) as a new solution to replace the aging ATS voice communication infrastructure. This paper conveys the industrial VoIP techniques for consideration.

1. BACKGROUND

The ATS voice service has been in service to coordinate air traffic transfer between adjacent FIRs. The ATS voice service is mostly based on a dedicated circuit using analog signaling such as Selective Signaling (SS-1/5), Dual Tone Multi-Frequency (DTMF), Automatic Ring Down, or Private Signaling System (PSS-1/QSIG).

Some States also use the voice/data multiplexer to compress the voice bandwidth into 8 kbps channel and then combine with Aeronautical Fixed Telecommunication Network (AFTN) data channel to transport over a 64 kbps dedicated circuit.

The use of voice/data multiplexer has allowed States to use lower the recurring cost of telecommunication service when the cost was very high in the 1990s. However, the voice/data multiplexer is mostly based on specific manufactories' proprietor technique that requires identical voice/data multiplexer on each site. This requires extensive coordination when the equipment need to be replaced or the equipment is no longer commercially available.

With the Internet explosion and advanced PC applications that use more and more bandwidth, the data network volume has increased dramatically and is now the dominant bandwidth consumer. This leads to recurring cost of telecommunication service becoming more affordable.

The VoIP has been used over public internet to provide long distance international calls with minimal cost to consumer. This is due to an open IP platform adopted by the industry.

The use of VoIP in ATC environment should be considered to reduce coordination and procuring of common platform requirement. The FAA is in the process to establish a test platform to evaluate VoIP integration to its existing ATS voice communication infrastructure and to its international ATS voice communication service.

2. DISCUSSION

2.1. Overview

This paper covers the voice over IP (VoIP) network for transporting voice and data over a common LAN or WAN infrastructure. This document discusses the major hurdles that need to be addressed when using either a LAN or WAN based VoIP network. There are 2 types of signaling conversion:

A. Translational Signaling

In Translational Signaling, the router/gateway actively participates in the call setup, connection, and tear down. It does this by locally emulating what would have normally been the remote equipment. This is shown in Figure 1 below

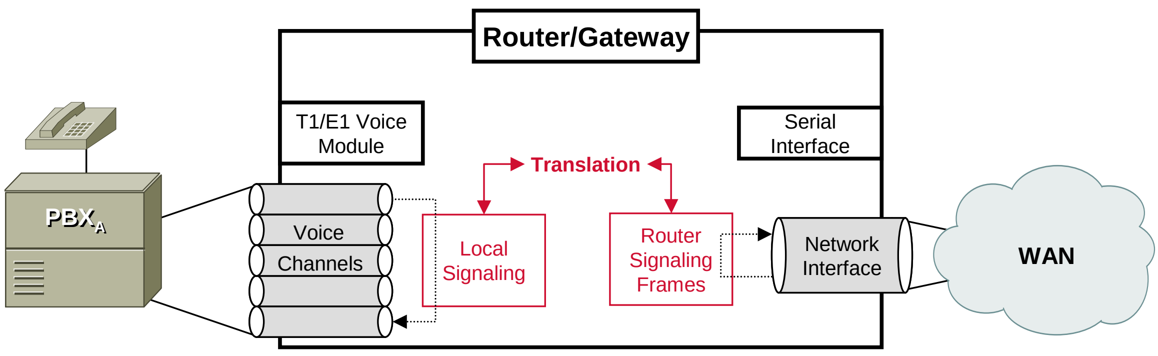


Figure 1: Translational Signaling Model

Notice that the local signaling is translated into a form the router/gateway network understands.

Another way to look at translational signaling is to look at where the 'signaling intelligence' is located in the network. To better illustrate, a typical tie line PBX network is shown in Figure 2.

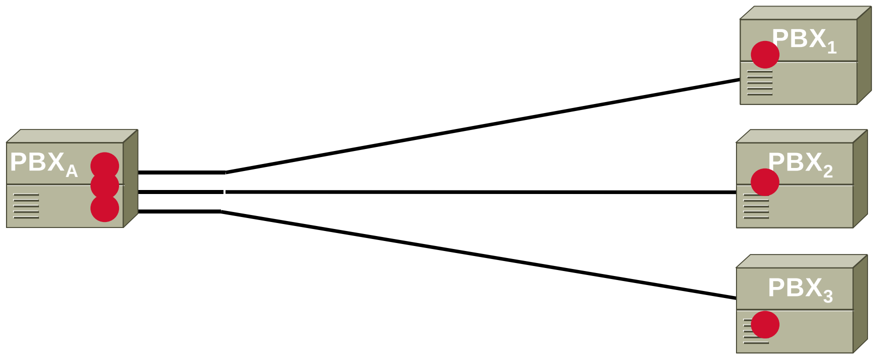
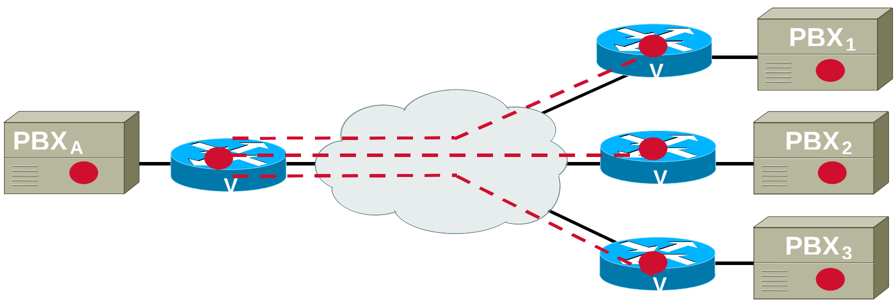


Figure Error! No text of specified style in document.: Location of Signaling Intelligence

The illustration describes a network of leased T1 lines between PBXs A, 1, 2, and 3. The red dots illustrate the locations of signaling intelligence, this is where the PBX indicates when and where the call will go. Note that there's signaling intelligence associate with each physical line.

Now consider the integrated network shown in Figure 3.



In this network, the signaling intelligence is distributed in the PBXs and in the router/gateways. This makes some elements of the network more complex, but it can save costs and improve performance.

B. Transparent Signaling

In a Transparent Signaling model, the signaling information is passed transparently through the router/gateway network and on through to the terminating PBX. In this model, all the voice channels are set up to be compressed at all times, and the PBXs never know the router/gateway network is operating. A diagram of how this accomplished is shown in Figure 4.

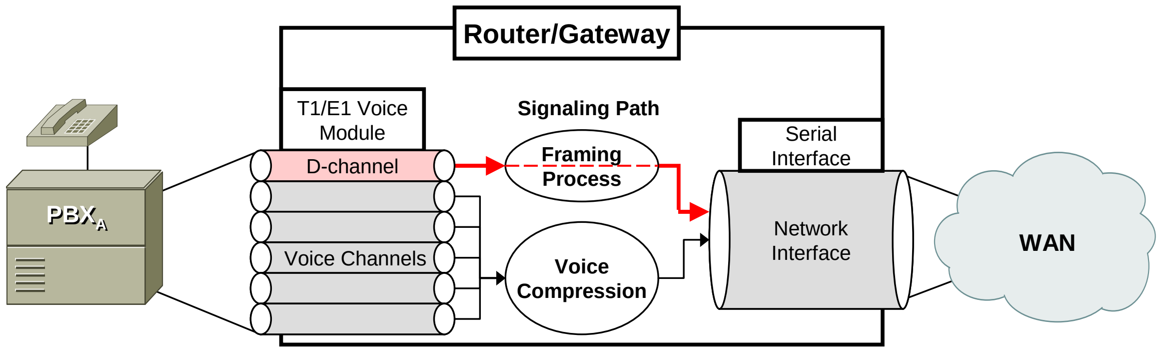


Figure 4: Transparent Model

As described in the figure, the signaling channel (d-channel) is framed independently of the voice channels, and frames are passed to the remote destination. Another way of looking at transparent signaling is shown in Figure 5.

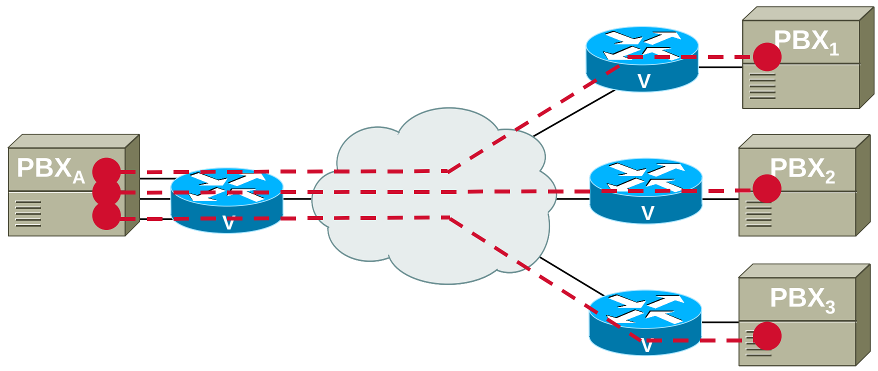


Figure 5: Signaling Intelligence in Transparent Model

Note that the signaling intelligence, represented by the red dots, stays in the PBX. The routers are dumb agents with respect to the signaling.

2.2 Signaling

Signaling can be the most difficult part of integrating a voice and data network. The most difficult part of using VoIP is not the technology but how to integrate existing ATC Voice Switching equipment signaling and the its IP network. Additionally, a lot of voice network support is outsourced, it's not always easy to know what's been configured and how it's supposed to work.

Signaling is how the telephone or PBX tells the phone network when and where to place the call. There are several different ways to break down telephony signaling that can accommodate different signaling types.

Analog	Digital
FXS/FXO • Two-Wire • Loop Start • Ground Start	CAS: Channel Associated Signaling • North American Robbed Bit • European, all bits in channel 16
E & M • Two-Wire • Four-Wire • Five Types (I, II, III, IV, V)	CCS: Common Channel Signaling • Out-of-band Signaling

Table 1: Signaling Summary

2.2.1 Analog Signal

Analog signaling is handled directly by the router/gateway. This means the router actively participates in the signaling for the call and therefore must be involved in the dial plan. In general, FXS interfaces are used to connect user telephones directly to the router/gateway, and FXO lines are used to connect the router/gateway to the PSTN network. E&M interfaces are used to connect the router/gateway to E&M trunk connections on the PBX.
All analog signaling is accommodated by translational signaling.

2.2.2 Digital Signal

2.2.2.1 Channel Associated Signaling (CAS)

CAS is also referred to as Robbed Bit Signaling. In this type of signaling, the least significant bit of information in a T1 signal is "robbed" from the channels that carry voice and is used to transmit framing and clocking information. This is sometimes called "in-band" signaling. CAS is a method of signaling each traffic channel rather than having a dedicated signaling channel (like ISDN). In other words, the signaling for a particular traffic circuit is permanently associated with that circuit. The most common forms of CAS signaling are loopstart, groundstart, Equal Access North American (EANA), and E&M. In addition to receiving and placing calls, CAS signaling also processes the receipt of Dialed Number Identification Service (DNIS) and automatic number identification (ANI) information, which is used to support authentication and other functions.
Each T1 channel carries a sequence of frames. These frames consist of 192 bits and an additional bit designated as the framing bit, for a total of 193 bits per frame. Super Frame (SF) groups twelve of these 193 bit frames together and designates the framing bits of the even numbered frames as signaling bits. CAS looks specifically at every sixth frame for the timeslot's or channel's associated signaling information. These bits are commonly referred to as A- and B-bits. Extended super frame (ESF), due to grouping the frames in sets of twenty-four, has four signaling bits per channel or timeslot. These occur in frames 6, 12, 18, and 24 and are called the A-, B-, C-, and D-bits respectively.

The biggest disadvantage of CAS signaling is its use of user bandwidth in order to perform signaling functions.

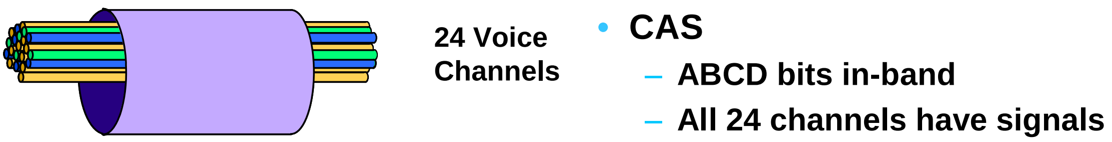


Figure 6: Channel Associated Signaling

2.2.2.2 Common Channel Signaling (CCS)

Another popular digital signaling method is Common Channel Signaling. In this signaling model one channel of the T1 or E1 circuit is set aside as a signaling channel. HDLC style messages are passed between endpoints to indicate on-hook and off-hook conditions, originating phone number, destination phone number, channel being used, etc., etc. ISDN signaling is a CCS protocol most often used to connect PBXs to the PSTN.

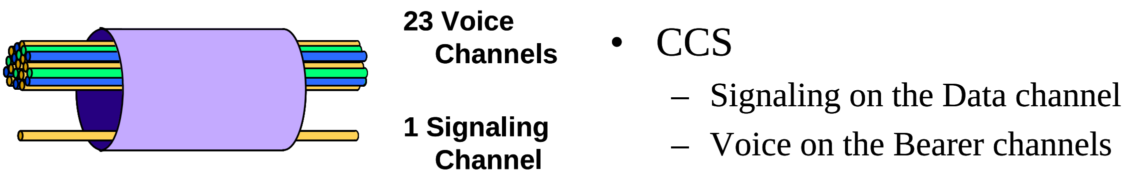


Figure 7: Channel Associated Signaling

Usually, CCS protocols for inter-PBX communication on a private network are proprietary, like Siemens' Corenet or Lucent's DCS+, and the vendors won't license the protocol to Cisco. Consequently transparent signaling is the only option in this scenario. The IOS feature for transparently passing the CCS d-Channel is called Transparent-CCS.

Qsig is an international standard protocol defining a CCS protocol that can be used to interconnect PBXs from different vendors while still maintaining many of the advanced features used on private PBX networks. Cisco equipment supports Qsig and can support translational signaling in voice networks that use it.

A network using Transparent CCS is shown in Figure 8

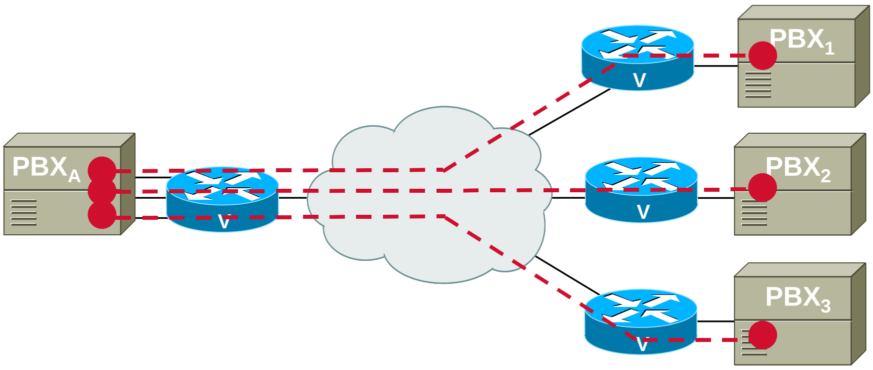


Figure 8: Transparent CCS

Note that with Transparent-CCS there is a one-to-one correspondence between PBX line card interfaces. Translational modes may reduce the need for line cards at the central cite and provide a more cost-efficient many-to-one relationship.

While Transparent-CCS is not as elegant a solution as the translation model using Qsig or CAS, this solution still has many advantages for the customer.

2.3 Voice over IP Bandwidth Requirement

Voice traffic uses a lot less bandwidth than traditional LAN-based computer networks. A single toll-quality phone call over the public network uses 64 Kbps in each direction – that's only 0.0625% of a 100 Mbps full duplex link. Each voice call takes up to 85.6 Kbps (64 Kbps + IP header + Ethernet header) in each direction supporting up to 1,160 calls over a full duplex link.

If bandwidth were the only issue, LAN-based IP telephony networks would have been deployed years ago. But other elements, such as bandwidth hungry business applications, advancements in telephone technology, and network congestion have been the major obstacles. Most of those issues have been resolved with newer VoIP technology.

2.4 Bandwidth and Compression

Just a few short years ago, the key driver for converged voice-data networks was saving on bandwidth. Today, with the reality of VoIP in the LAN, the new driver is laying the groundwork for hybrid applications like call centers and unified messaging.

This doesn't mean that bandwidth is no longer an issue, in fact, until bandwidth is free, voice compression, its characteristics, and its limitations will continue to be issues for network engineers. Voice compression is usually done in the manner shown in the Figure 9

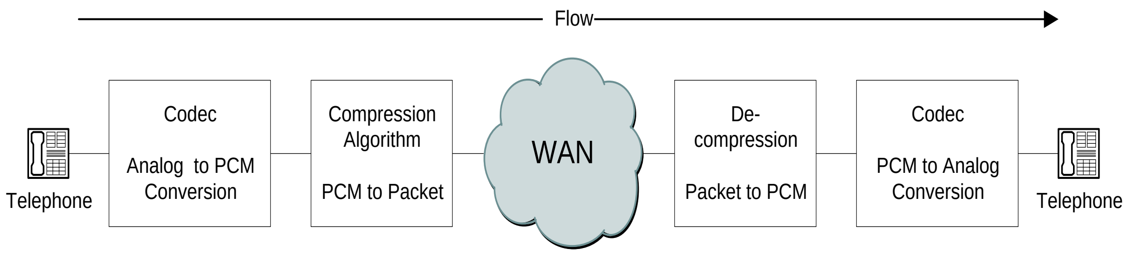


Figure 9: Voice Compression Flow Diagram

The analog signal from the telephone is digitized into PCM signals by the voice CODEC. The PCM samples are then passed to the compression algorithm which compresses the voice and loads it into a packet format for transmission across the WAN. On the far side of the cloud the exact same functions are performed in reverse order.

Depending on how the network is configured, a router/gateway can perform both the CODEC and compression functions or only one of them. For example, if an analog interfaces is used on the router/gateway, then the router/gateway performs the CODEC function and the compression function as shown in Figure 10

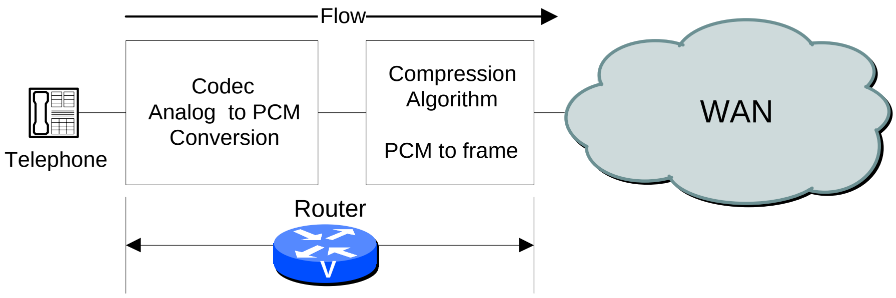


Figure 10: CODEC function in Router/Gateway

If instead, a digital PBX is used, the PBX performs the codec function, and the router/gateway just processes the PCM samples from the PBX. An example is shown in Figure 11

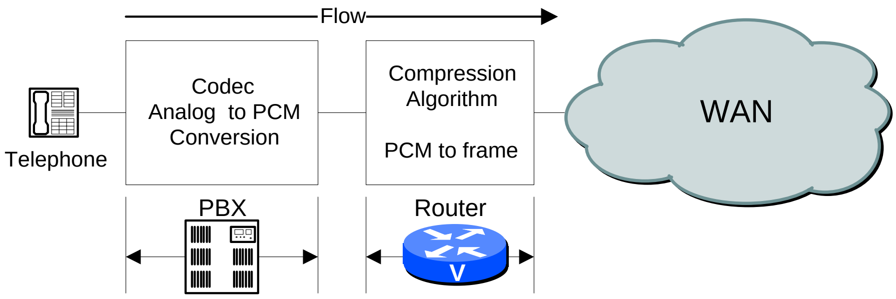


Figure 11: CODEC function in PBX

2.5 Compression Algorithms

Table 2 lists common compression algorithms and their nominal bandwidths.

Coder	Nominal Bandwidth
PCM G.711	64 Kbps
G.726 ADPCM	32 Kbps
G.726 ADPCM	24 Kbps
G.726 ADPCM	16 Kbps
G.728 LD-CELP	16 Kbps
CS-ACELP G.729	8 Kbps
CS-ACELP G.729A	8 Kbps
CS-ACELP G.729B	8 Kbps
CS-ACELP G.729AB	8 Kbps
G.723.1 MP-MLQ	6.3 Kbps
G.723.1 ACELP	5.3 Kbps
G.723.1A4 MP-MLQ	6.3 Kbps
G.723.1A4 ACELP	5.3 Kbps

Table 2: Considered Voice Compression

The ‘Nominal bandwidth’ is how much bandwidth the voice stream requires if it were on the wire by itself, without packet headers or flags. Technically, PCM is an encoding method and not a compression algorithm. As noted in the previous section, the compression algorithms require PCM streams as the input format.

2.6 Required Bandwidth

The simplest way to calculate the per-call required bandwidth is to determine the overhead factor and multiply it by the bandwidth. The overhead factor is simply the total number of bytes used to transport the voice payload (including the inter-frame flag) divided by the number of voice payload bytes.

$$\text{Required Bandwidth} = \frac{(\text{Payload + Header + Trailer + Flag})}{(\text{Payload Size})} \text{ X (Coder Bandwidth)}$$

Notice how the ‘byte’ units in the overhead factor cancel out, leaving you with bandwidth on both sides of the equation. For example, a Frame Relay frame with 30 bytes of G.729 payload. The calculation would look like this:

1 ea Flag Byte
2 ea Frame Relay Header Bytes
3 ea Voice Header Bytes
30 ea Voice Payload Bytes
<u>2 ea CRC bytes</u>
38 Bytes Total

Therefore, to transport 30 bytes of voice payload, a 38 bytes of information is needed. This gives an overhead factor of 38/30 = 1.267.

By multiply the voice bandwidth by the overhead factor to get the total per-call voice bandwidth required. In example the required bandwidth is (8 k) x (38/30) = 10.13 Kbps.
This formula works for all compression algorithms if the payload size and overhead figures are defined. Any byte that’s not in the voice payload that’s used to send the frame should be considered overhead.

2.7 VoIP Header Compression

Header Compression is used in VoIP to reduce the overhead on low-speed lines. An ordinary VoIP header contains 20 bytes of IP header, 8 bytes of UDP header, and 12 bytes of RTP header. Add this to a 2 byte data link header, a 2 byte data link CRC, and a flag, and the result would be 45 bytes of overhead relative to a 20 byte CS-ACELP voice payload. This results an overhead factor of 3.25 and a per-call required bandwidth of 26 Kbps.

Using header compression, the total header can be reduced to 2 or 4 bytes. This gives an overhead factor of 1.45 and reduces the required bandwidth used to just 11.6 Kbps per connection.

However, as discussed previously, IP Header compression drains CPU resources so care must be taken if a lot of calls are to be terminated.

3. CONCLUSION

The meeting is invited to note the VoIP techniques for consideration in the future replacement of existing ATS voice telecommunication service as well as integrating the voice service into IP network.

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